S1000 V2 manual

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professional

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AKAI VERSION 2.0 SOFTWARE FOR THE S1000 STEREO DIGITAL SAMPLER

INTRODUCTION

It's fair to say that the S1000 has become something of an industry standard as far as 16-bit stereo samplers are concerned having outsold every other sampler and most professional musicians (as well as many semi-professional and 'amateur' musicians) can be seen with one in their racks. Similarly, the S1000 has been embraced by the audio visual world where it's high quality sampling is used for post-production. Recording studios now have one (or sometimes more) as their main sampler for general keyboard work and for production effects and many famous artists and studios who once used very expensive sampling systems have now gone over to the S1000.

As a result of all this, we have had extensive feedback about the features you want to see in the instrument. To that effect, we have responded with this, the Version 2.00 update for the S1000 which contains many new features that include:

1. Timestretch (*)
2. Resampling at a variable bandwidth (*)
3. Sample merging (*)
4. Level control of samples in sample edit (*)
5. Loop tuning (*)
6. The ability to read programs as well as samples off S900/S950 disks
7. Velocity sample start
8. Variable velocity sensitivity for individual keygroups
9. Selected programs are now placed top of screen in SELECT PROG
10. Keygroup mapping is now done using keyboard input
11. Invertable filter keyboard tracking
12. Individual program tuning
13. Improved delete procedures
14. Improved programming procedures and defaults
15. Keyboard position control of LFO speed, depth and delay
16. Automatic hard disk load via MIDI program change
17. You can back up the contents of a hard disk or the internal memory to RDAT via the IB104 digital i/o interface option
19. Sampling can be done digitally using he IB104 digital interface for improved sample quality with up to 48kHz sample rate (*)
20. Sample data can be sent out over SCSI via the IB103 SCSI interface to facilitate faster sample dumping to computers for editing, etc.
21. The ability to use hard disks greater than 60 Meg such as Sony optical drive

The new software is available free of charge on disk but there will be a small charge to have it installed on chip. You may freely copy the software. To copy the software, simply insert a blank formatted disk (an ordinary DD will do), go to the DISK page and then, by moving the cursor to the type of data you want to save, select OPERATING SYSTEM. Press SAVE and then press F8. The S1000 will prompt you as saving the operating system will wipe the entire memory. If you are happy to proceed, press F7. The software is now saved to disk and can be used simply by inserting the software disk in the disk drive and turning the S1000 on. *(Note: owners of an S1000 with a hard disk can save the new operating system to the first volume of the hard disk and this will automatically load up when you switch the S1000 on.)*

* NOTE - Owners of an S1000PB which does not sample, will not be able to use any of these sampling or sample editing functions.

CONTENTS

1. NEW PROGRAMMING TEMPLATES AND DEFAULTS

2. EDIT SAMPLE: Timestretch, Resampling, Sample Merging, Sample Level Control, Loop Tuning.

3. EDIT PROGRAM: Velocity Sample Start Time, Program master tune, Individual Keygroup Velocity, Filter Key Follow, New LFO options, Setting Keyspan using MIDI keyboard, New SMP1 display, Changing keygroups levels.

4. DISK: Assigning automatic hard disk loads using MIDI program change, Formatting hard disks larger than 60Meg, Loading S900 disks, New delete options, Saving SCSI ID numbers.

5. MIDI: New MIDI Receive display.

6. MASTER TUNE: New soft pedal options

7. TIPS + HINTS: Using the individual outputs, Drum sampling, Sample start time editing.

NEW PROGRAMMING PROCEDURES AND DEFAULTS

Since the S1000 was introduced, we have found that certain defaults are not as well chosen as they could have been and our programmers and also our customers have suggested ideas that would assist in programming and sampling, especially from scratch. The new defaults for Version 2.00 in Edit Sample are as follows:

The synth waveforms are not now included. It was found that no-one used these waveforms as the basis of a sound (the original intention was so that the S1000 could be used as a basic synth as well as a sampler). As a result, certain aspects of EDIT PROGRAM are easier as we shall see soon.

In Edit Sample, as no samples are present in the S1000 when you power on with no sound disk inserted, when you make your first sample the S1000 automatically assigns the name NEW SAMPLE. You may change this using RENAME, of course. Nothing has changed in REC1 but in REC2, the meter is automatically switched to ON so that you eliminate one level of button pushing. Should you need to turn this off, simply hit F7 (OFF). The rest of REC2 is unchanged.

Now, when you play a 'raw' sample in EDIT SAMPLE, there is not the long, sloppy release when the sample is played. It now stops instantly.

When you go to trim a new sample (ED1 - F4), the display defaults to fully zoomed out so that you can see the whole of your new sample. If you wish to look more closely at it, press F5 (ZIN) repeatedly until the required magnification is reached.

Other defaults exist in the timestretch and resampling pages. Timestretch is set to INTELLIGENT mode with QUAL and WIDTH defaulting to 90 respectively. This gives excellent timestretch results with very little effort on your part - all you have to do is set the stretch zone, the stretch factor, give the sample a new name and press GO!! The resampling page defaults so that your new bandwidth is set at half the original bandwidt. There are also two new soft keys marked 2/3 and 3/4 which automatically resample at those fractions of the original bandwidth. This allows you to save quite a bit of memory but not have too adverse an effect on the sounds quality. For more detailed information on timestretch and resampling, please refer to the relevant sections of this manual.

Other than those, no other changes have been made to default parameters in Edit Sample.

The defaults that exist within Edit Program have been changed more extensively.

To begin with, the envelopes (both ENV1 and ENV2) now have shorter release times to prevent the 'sloppy' after-ring that samples used to have in the old software. Also, ENV2 defaults to a more basic shape which is easier to work from. The filter page now has keyboard tracking set to 12. With the filter cutoff at 99 this will have no effect but if you set the cutoff lower, you will have keyboard tracking ensuring an even tone over the keyboard range.

In the PMOD page (now renamed to PTCH) the default for pitchbend is the more commonly used 2 semitones (i.e. 1 tone). The LFO is also programmed slightly differently with a more 'musical' vibrato speed. LFO delay is not now set as this was felt to be more useful. All other parameters on this page remain the same.

In the OUT page you will note that the basic loudness has been changed to 80 instead of 90. This is to give more control over turning individual keygroups up in level. In the old software with this parameter set to 90, there was no headroom for turning keygroups up in SMP2 - now you will notice more effect when you try to increase the level of keygroups. *(Note: you may find that you will need to tweak either the output level of your S1000 up or will need to adjust the gains of you mixer to make up for this change but this is only slight and will not affect the signal to noise ratio.)* Still in the OUT page, you will note that effect send is now set to 'OFF' and that individual output assignment is also set to 'OFF'. Also, the DELAY parameter of the PAN section has been changed to read 00. Other than those changes, everything else remains the same.

The most significant change is in SMP1. Because the synth waveforms are no longer present in the S1000 when you switch it on, you don't have the confusing display of waveforms in the four velocity zones. Instead, only zone 1 has a sample in it which makes things a lot clearer when you are trying to assign samples to keygroups.

No other changes have been made to Edit Program.

In SELECT PROG, Version 2.00 places the selected program at the top of the screen when you dial up a new program number. This does two things. Firstly, it allows you to see exactly which program you are playing and, secondly, it ensures that the program you have currently selected is the one selected for editing when you hit Edit Program (on the old software, it was possible to enter Edit Program to work on the program you were playing and be editing the wrong program!!).

NEW DELETE FUNCTIONS

The Version 2.00 software offers new, safer delete functions. On the old software, it was all too easy to lose valuable data, either program or sample and so failsafes have been built in.

When you try and delete a program in Select Program you are now asked "delete one program?" to make sure you really want to. If you do, press F7, if you don't, press F8. Then the S1000 scans the other programs to see if the samples contained within the program you wish to delete are used elsewhere. If they are, then nothing else happens but if the samples contained in the deleted program are not used anywhere else, the S1000 asks you "delete XX released samples?" where XX relates to the number of samples in that program not used elsewhere. If you wish to proceed and delete these samples, press F7 but if you don't, press F8. In this way, you can delete both programs AND samples simultaneously.

When you try and delete individual samples in Edit Sample, you are asked "delete one sample" - press F7 to do so or F8 if you change your mind. Similarly, in Edit Program, if you try an delete a program, you are asked "delete one program?". Again, F7 will get rid of that program, F8 allows you to abort the deletion. As in Select Program, this deletion offers the option of deleting the samples contained within that program if they are not used in any other programs.

Should you wish to delete anything from disk, either floppy or hard, the same failsafes apply and the S1000 checks that you really wish to proceed. Now there is no excuse for losing valable data!!

NEW EDITING FACILITIES IN EDIT SAMPLE

TIMESTRETCH

One of the most powerful new facilities on the S1000 is TIMESTRETCH. This unique method of sample processing allows you to stretch or shrink a sample without changing its pitch, a facility that is only available on very expensive computer based sampling systems and even then sometimes as an expensive update or option!!. There are many applications of this process of which the most obvious is probably lengthening or shortening a sampled drum loop so that it fits tempo-wise with the rest of the track but so that its pitch isn't altered. There are, however, many other uses.

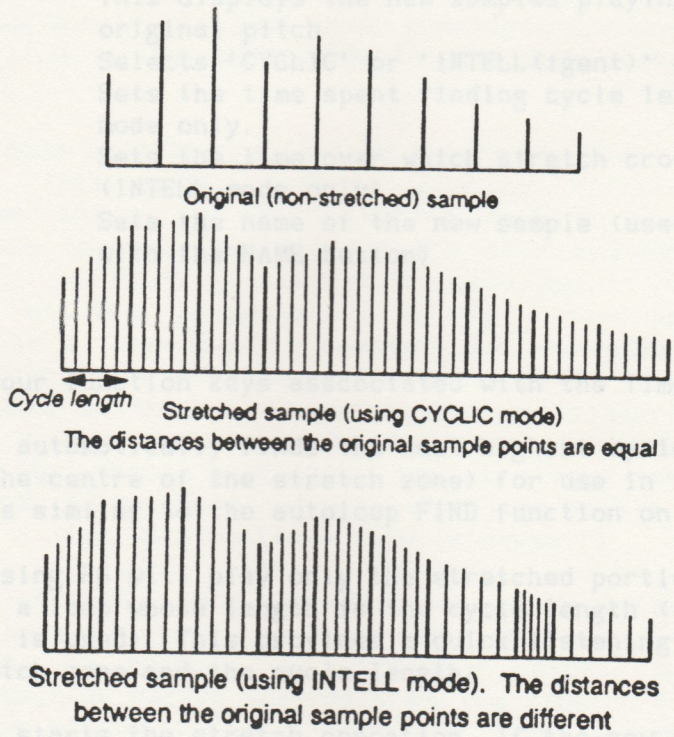
For example, it is common to sample a sound where it is only possible to sample one note (for example, a blown bottle). This usually produces acceptable results when the sound is played across the keyboard but, in cases where the sound does not transpose well you could use Timestretch to create a multi-sample of that one sound. This can be especially useful on something like, for instance, a cello. You could have a situation where you had one marvellous cello sample that captured the quality and character of the cello beautifully but that the other samples you took were not quite so good. You may like to try using that one sample on its own mapping it out across the entire keyboard. The problem is, however, that as you play that one sample higher up on the keyboard, the natural vibrato of the cello speeds up and sounds odd. Using Timestretch, you could process the sample creating a multi-sample of that one cello sample where the vibrato stays more or less constant across the keyboard range.

Other uses for it include A/V work where whole music tracks could be lengthened or shortened. This can be particularly useful for making a piece of music fit a music cue more precisely or it could be used so that a jingle fits precisely into an advertising slot on TV or radio. There are many uses for it in film and video dubbing where a voice-over could similarly be made to fit more precisely. In a recording studio, there are many applications for timestretch especially in the technique of 'spinning in' backing vocals or guitar solos or whatever.

As you can see, there are many creative possibilities using Timestretch and as you become acquainted with its operation, you will find many more uses for it. But how does it work?

TIMESTRETCH - THE THEORY

When you stretch a sample, you are effectively moving the gaps between the sample points further apart. This would ordinarily have the effect of the sample being lower in pitch. With timestretch, the S1000 inserts bits of the original sample data in between these gaps and so keeps the pitch the same but makes the sample longer. When a sample is shrunk (i.e. made shorter), the S1000's processor removes bits of sample data from the original sample thereby making it the same pitch but shorter in length. How the S1000 actually does this is unique to Akai and enables the S1000 to produce high quality stretched or shrunk samples that are indistinguishable from the original sample. You also have a good deal of control over the end result as we shall see a bit later. In practice, you need not worry too much exactly what is going on the S1000 while it is performing this amazing operation and timestretching can be a straightforward and rewarding sampling process.


Original (non-stretched) sample
Cycle length
Stretched sample (using CYCLIC mode)
The distances between the original sample points are equal
Stretched sample (using INTELL mode). The distances between the original sample points are different

USING TIMESTRETCH

Timestretch is available in ED2 of EDIT SAMPLE. To process a sample, go to EDIT SAMPLE and press ED2 and you will see that F3 now has the new parameter TIME. Press F3 and the screen will display the Timestretch parameters.

The parameters are as follows:

SAMPLE:	This selects the sample to be stretched/shrunk
STRETCH ZONE....TO	This sets the portion of the sample you wish to stretch/shrink
CYCLE LENGTH	Sets the sample period in the 'CYCLIC' mode (see below)
TIME FACTOR	Sets the amount of stretch/shrink (100% = no change)
TOTAL	This displays the new samples length in points and by percentage of memory
NORM. TIME	This displays the new samples playing time at its original pitch
STRETCH MODE	Selects 'CYCLIC' or 'INTELL(igent)' modes
QUAL(ity)	Sets the time spent finding cycle lengths in INTELL mode only.
WIDTH	Sets the time over which stretch crossfades are made (INTELL mode only)
NEW SAMPLE	Sets the name of the new sample (used in conjunction with the NAME button)

There are also four function keys asscociated with the Timestretch page.

F5 = AutC - This automatically finds the next highest cycle length (based on the centre of the stretch zone) for use in the cyclic mode. It is similar to the autoloop FIND function on the loop page.

F6 = ZONE Pressing F6 will play only the stretched portion of the sound with a loop whose length is the cycle length (if the cyclic mode is used). This provides a quick listening test for the stretch zone and the cycle length.

F7 = GO This starts the stretch operation. If the new sample already exists (i.e. if you have already tried a stretch), it is first automatically erased. During operation, you can abort the process by holding down F8 for a few seconds. If aborted, the new sample will not be created.

F8 = PLAY This plays the resulting new sample.

To stretch a sample, here's what to do.

Assuming that you are in EDIT SAMPLE, press ED2 and then TIME. Select the sample you wish to stretch by scrolling using the data wheel. When you have done this, press NAME and name the new sample in the usual way giving it a slightly different name to the original (you could, perhaps, insert a '+' in the name to indicate a stretched sample or a '-' to indicate a shrunk sample). Now press ENT. If you wish to stretch (or shrink) the entire sample, set the STRETCH ZONE parameters so that they equal the length of the original sample. For example, if the original sample is 0-34870, then set the STRETCH ZONE to that length (note: the stretch zone is always one figure lower than the total sample length - in this example, the maximum stretch zone you could have would be 0-34869).

Sometimes, it is desirable, especially on single instrumental samples, to stretch just the sustain portion of the sound. To do this, set the STRETCH ZONE parameters accordingly.

Before we move on any further, let us first discuss the two stretch modes in more detail.

There are two modes you can use for timestretch, CYCLIC and INTELLIGENT. Cyclic is better suited to single pitched instruments such as individual instrumental samples. Intelligent is best for speech and music processing. In the cyclic mode, there is a parameter known as CYCLE LENGTH. The best setting for this is when it matches the loop length of the original sample. If you have trouble finding a satisfactory cycle length, the S1000 will do it for you when you press F5 (AutC). A steady part of the sample (i.e. somewhere after the initial attack transient has been completed) can often be stretched by as much as 1000% producing a clean result which contains the same 'life' as the original but spread over a longer period of time. The rate of operation (i.e. the length of time it takes for the S1000 to calculate the new sample) is just over 2 seconds for every second of the original sample (i.e. a four second sample will take just over 8 seconds to process).

The Intelligent mode is better suited to speech and music. To get technical for a moment, this mode performs special correllation tests to automatically set the cycle length before each part of the sample is stretched. Because these correllation tests take a long to time for the S1000 to perform, a QUAL(ity) control is provided to set the degree of these tests. If the QUAL parameter is set to a high value, the stretch operation can take a long time to complete but acceptable results can be obtained using fairly low settings. Your best bet is simply to experiment with different QUAL settings. In most cases, particularly if the stretch factor is high, a poor result can be improved by starting again with a higher QUAL setting. (Note: execution time is always longer than for the CYCLIC mode and can take several minutes if a long sample is stretched by a high factor with a high QUAL setting. This is normal.)

When you select the INTELL mode, an additional control 'WIDTH' is included. This sets the time over which the original sample data is crossfaded with the inserted data to produce the stretched result. Without the width control, the crossfade varies according to the stretch factor - it is low for large amounts of stretch or shrink and high for small amounts. The width control actually limits the maximum crossfade width which will be used.

If you decide to use the CYCLIC mode, you may now select the best CYCLE LENGTH either by adjusting this parameter manually or by pressing F5 which will automatically find the best cycle length for you. You can now select the amount by which you want to stretch or shrink the sample. To shrink a sample, rotate the data entry wheel counter clockwise so that the percentage is less than 100% and to stretch a sample, move the data entry wheel clockwise to enter a value greater than 100%. The total range is 25% (shrinking) to 2000% (stretching). When you do this, the TOTAL and NORM. TIME fields will change to display the new samples length and playing time.

If you choose to use the intelligent mode, the cycle length parameters are not applicable and you must set the QUAL and WIDTH controls. In practice, the best results are achieved when a high quality setting is used. This ensures that plenty of sample data is inserted and ensures a smooth result. Using a high quality setting will, however, take a long time to process so

you may choose to use a lower quality setting. There are no rules here except that higher quality setting produce better results. The width control crossfades the original data with the inserted data. Generally, the rule is to use low width settings with high quality settings and high width settings with low quality settings although you may have to experiment to get the best results.

(Note: the processing time in intelligent mode can take a very long time. A four second sample with a stretch factor of, say, 125% and a high quality setting can take several minutes. There is nothing wrong with your S1000 if it seems to be taking a very long time and it hasn't 'crashed'. If you wish to stop the processing, hold down F8 for a few seconds.)

Regardless of which mode you have selected, once you have set everything simply press F7 (GO) and the S1000 will begin its calculations. When you do this, the 'new sample' status display is replaced with a TIME TO GO xxx% display to show the progress of the stretching. When the process is complete you can hear the results of your work either by pressing F8 (PLAY) or by hitting SLCT (F1) and selecting the new sample in the usual way. You may now treat the new stretched or shrunk sample just like any other sample and use it in programs, saving it to disk, etc...

Note 1: The new stretched or shrunk sample will lose the loop(s) set in the original. You will have to set up a new loop in the processed sample.

Note 2: Because you have to make a copy of the original for the timestretch process, you may find that you don't have enough memory to accommodate both the original and the processed sample (the display will tell you). The only way around this is to delete one or more samples to make room for the new one (make sure they are saved to disk first, though!!). If you are trying to stretch a long sample such as backing vocals or a guitar solo or whatever, something that uses the entire 24 second memory of the S1000, the only course of action available to you is to use the EXM005 memory expansion boards.

To recap then, to stretch or shrink a sample, use the following procedure:

1. Go into ED2 of EDIT SAMPLE and select TIME. Select sample to be processed.
2. Make a copy of that sample using the NAME function.
3. Set the stretch zone (i.e. which part of the sample will be processed).
4. Select either CYCLIC or INTELLIGENT mode as appropriate.
5. If cyclic is selected, set cycle length - if intelligent is used, set the QUAL and WIDTH parameters accordingly.
6. Set the stretch factor - i.e. the amount you wish to stretch or shrink the sample.
7. Press F7 to initiate processing.
8. When complete, press F8 (PLAY) or press F1 (SLCT) and audition the samples in the usual way.
9. Set up a new loop (or series of loops) in the new sample if necessary.

It must be pointed out that timestretch, like looping, is not always a perfect technique. It is often a compromise but in most cases, extremely good results can be obtained that are tonally indistinguishable from the original but you may need to experiment with some sounds to achieve precisely the right effect - sometimes it will be impossible to get a perfect result without some imperfections. The defaults have been very well chosen for optimum results with most samples so there should be little button pushing and data entry to be done so you should be able to get good results quite quickly and effortlessly. Having said that, however, you may need to practice to become an expert timestretcher!!

TIME STRETCH PERCENTAGE - CENTS TRANSPOSITION TABLE

TRANSPOSITION	TIME FACTOR PERCENTAGE
UP 1 semitone	106%
UP 2 semitones (1 tone)	112%
UP 3 semitones (minor 3rd)	119%
UP 4 semitones (major 3rd)	126%
UP 5 semitones (major 4th)	133%
UP 6 semitones	141%
UP 7 semitones (major 5th)	150%
UP 8 semitones	159%
UP 9 semitones	168%
UP 10 semitones	178%
UP 11 semitones	188%
UP 12 semitones (octave)	200%
DOWN 1 semitone	94%
DOWN 2 semitones (1 tone)	89%
DOWN 3 semitones	84%
DOWN 4 semitones	79%
DOWN 5 semitones (major 4th)	75%
DOWN 6 semitones	71%
DOWN 7 semitones (major 5th)	67%
DOWN 8 semitones	63%
DOWN 9 semitones	60%
DOWN 10 semitones	56%
DOWN 11 semitones	53%
DOWN 12 semitones	50%

NOTE: all percentages have been rounded

The transposition formula is:

To transpose upwards, multiply 1.0595 by itself for as many semitones and multiply the result by 100. i.e. for three semitones -

1.0595 X 1.0595 X 1.0595 X 100 = 119%

To transpose a sample downwards, divide 100 by 1.0595 multiplied by as many semitones is required. i.e. for a transposition down of one tone -

100(1.0595 X 1.0595) = 89%

RESAMPLING

One of the most useful features on the Akai S900 and S950 samplers is the ability to sample at a totally variable bandwidth. This allows you to optimise bandwidth according to pitch and tone and also save tremendous amounts of memory. Most samplers only offer a few switched bandwidths which is not ideal. Unfortunately, owing to a hardware limitation, the S1000 is not able to select a totally variable bandwidth when sampling so the solution to this is to resample afterwards at a totally variable bandwidth. Not only can you optimise your sample rate against pitch and the instruments tonal characteristics but you can save yourself lots of memory making room for even more samples.

The resampling functions are available in ED2 of Edit Sample by pressing F4 (RATE). The display shows the selected samples original sampling rate in Hz and there is a field directly below this that allows you to set the samples new bandwidth. The other fields simply show how much memory the new sample will take at its new bandwidth and what tuning offset has been used in the resampling process but more on that a bit later.

As with timestretch, you have to make a copy of the selected sample to create the new, resampled sample. This makes a lot of sense because if resampling was totally destructive (i.e. you overwrote the original) and you didn't like the result, you might not be too pleased. The only way to overcome this would be to save the original sample to disk first which would be far too time consuming, especially in a studio where the clock is ticking away!!

So, to make a copy of the selected sample, press NAME and give the new sample a different name. You may like to include a clue to indicate that it is a resampled sample - perhaps put an 'R' after the name if you have room. Having done that, press ENT to enter the name. Now move the cursor to the field marked 'New Sample Rate' and set the bandwidth accordingly. This field defaults to exactly half the original bandwidth so you can use that if you wish although you can change it to any value you wish. Furthermore, there are two function keys marked 3/4 and 2/3 and these automatically set the resampling to three quarters and two thirds of the original bandwidth. Pressing either of these keys or GO (F7) will initiate the resampling process. To hear the results of this simply press F8 (PLAY) which will play the new sample at its new bandwidth. You should find that there is little or no audible tonal change and no extra digital noise or clock interference.

The reason for this is that highly efficient filters switch in to ensure that no digital noise is introduced to the signal. The default is 'AutoF' where the filter is automatically set at the optimum frequency according to the samples new bandwidth. There are other options, 'Filt1' through to 'Filt4' and these simply offer a means to set your own filter rate with varying degrees of effect. Filt1 has a high filter frequency, Filt2 is slightly lower, Filt3 lower again and Filt4 being the lowest filter frequency option. In practice, you will find that AutoF produces the best results with little or no effort on your part and so it is recommended that you stick with that but don't be afraid to experiment with the other settings.

The only other fields not explained so far are the 'new length' and 'tune offset' fields. The 'new length' field shows the size of the new sample at its new bandwidth in sample points and as a percentage. This will give you a good indication of how much memory the new sample will use up. The 'tune offset' field shows the amount by which the new sample will be transposed to accommodate the new bandwidth. Whenever a samples bandwidth is changed, its base pitch has to change. In order for this not to affect the actual pitch of the sample, a tuning offset is introduced to keep it at the same pitch relative to the key it was sampled on. For example, if you resample a sample taken at C3 at exactly half bandwidth, without the tuning offset, that sample, when played on C3, would sound an octave too high. The tuning offset pitches the sample an octave down (i.e. -12 semitones) so that the sample is at the correct pitch when you play C3. Without this offset, you would have to either retune every new sample or edit every program that uses the new sample which is something you would probably not be too pleased about!! Neither of these fields, 'new length' or 'tune offset' are available to be programmed by you so you needn't worry at all about them. All you need to do to set a new bandwidth is select the sample you wish to process, copy it, set the new sampling rate and press GO - couldn't be simpler. After that, you can delete the original if you wish to make room for more samples.

Note 1: When you resample something and give it a new name and then delete the original, if that original sample already exists within a program, you will find that you will not hear anything when you try to play the program. If you look at SMP1 in Edit Program, you will see that there is a '?' alongside the original samples name. What has happened is that the program is still set up for the old samples name that you have now deleted and it can't find it in memory (hence the '?'). You must now assign the resampled sample to that keygroup. Alternatively, you could rename the resampled sample to its original name now that the original sample has gone and this will place that sample in the program automatically.

Note 2: You may find that the loop in your new sample glitches slightly. This is not a problem and you can usually get a smooth loop back simply by going to the LOOP page and pressing F7 (FIND) which will activate the autoloop function and the S1000 will find a new smooth loop point for you.

Note 3: Because you have to copy a sample to set a new bandwidth, you may find that you don't have enough memory to accommodate the new sample (the display will tell you). You will have to delete one or more samples to make room for this. Just make sure those other samples are saved to disk first, however!!

Note 4: Don't forget that you can resample a resampled sample to make even more memory available!!

OPTIMUM BANDWIDTH CHART

INSTRUMENT	OPT B/W	INSTRUMENT	OPT B/W	INSTRUMENT	OPT B/W
Kick drum	10k	Bright snare	16-18k	Crash cymb	17-19k
Clicky kick	12k	Woody snare	14-16k	Ride cymbal	15-17k
Round kick	8-10k	Round snare	14-15k	Splash cymb	17-19k
Gated kick	10-13k	Gated snare	15-18k	Hi-hats	16-19k
Toms	13k	Congas	13-15k	Synth Snare	14-16k
Deep toms	11-12k	Timbales	14-16k	Synth Kick	10-13k
Stick toms	13-15k	Octabans	12-14k	Synth Toms	12-15k
Cabasa	14-16k	Cowbell	12-15k	Rimshot	12-14k
Claves	12-14k	Agogo bell	13-14k	Side stick	12-14k
Claps	12-15k	Vibraslap	14k	Brush Snare	12-14k
Maracas	13-14k	Whistle	13k	XXXXXXXXXXXXXXXXXX	
Bass guitar	10-12k	Acoust guit	12-14k	Piano	14-16k
Slapped bass	13-15k	Nylon guitar	10-14k	Rhodes pno	12-14k
Synth bass	12-15k	Elect guitar	12-15k	FM Rhodes	14-16k
Upright bass	10-12k	Distorted gt	13-15k	Clavinet	12-16k
Fretless	10-12k	12 str guit	13-16k	Organ	12-16k
Violin	14-16k	Oboe	14-16k	Trumpet	14-17k
Viola	12-14k	Clarinet	14-16k	Muted trump	15-18k
Cello	11-13k	Bassoon	12-14k	French horn	13-15k
Contrabass	10-12k	Flute	12-14k	Trombone	12-16k
Soprano sax	14-16k	Alto sax	13-15k	Baritone sx	12-14k
Vibraphone	16-18k	Tubular bell	15-17k	Voice	12-14k
Marimba	14-16k	Gong	13-15k	Mixed choir	12-14k
Xylophone	14-15k	Tympani	12-14k	Female chr	12-14k
Glock/Celest	15-16k	Triangle	15-18k	Breathy vox	12-14k
Digi synth	14-19k	Analog synth	12-15k	Orch strike	14-19k
String sect	14-16k	Brass sect	14-17k	Sax section	14-16k

You will note that these bandwidth figures are approximate. Of course, you may require higher or lower bandwidths than those shown for certain instruments - it all depends entirely on the tone of the instrument you are sampling. If it is bright, it may require a higher bandwidth than the one recommended but if the sound is relatively mellow, you might like to try a lower bandwidth setting. The best thing to do is experiment. Resample at your selected bandwidth and play it back from the MIDI controller comparing it with the original. If it sounds at all muffled, try a higher bandwidth but if it sounds good, keep it and delete the original if you wish to clear up memory. If it sounds o.k., you might like to try a slightly lower bandwidth.

SAMPLE MERGING

As well as being able to splice two samples together and crossfade them over a period of time, you can now merge two samples together. This opens up new possibilities for layered sounds but without the resultant loss of polyphony you get if you layer samples in keygroups within a program. For example, using sample merging you can layer brass on top of strings to create rich orchestral textures. You can stack two similar samples on top of each other for detune effects or you could layer various synth sounds on top of each other for huge, fat electronic textures. Of course, you could layer, for example, rich analogue synth strings on top of detailed string samples to create perhaps the ultimate sampled string sound. But how is this achieved?

Sample merge exists within the old JOIN page which now looks decidedly different with the new software. Many attributes from the old software still exist and you may be familiar with them already. Function buttons F7 and F8 have changed and there is also a new set of fields that allow you to adjust the relative levels of the two samples so that they can be mixed perfectly.

To merge two samples together, press F4 (ED1) and then F4 again (JOIN). Move the cursor to the field marked 'A :' and scroll through the samples until you find one of the samples you wish to merge. Move the cursor to the next field marked 'B :" and scroll through until you find the other sample you wish to merge. Once you have done that, move the cursor to the field marked "J :", press NAME to create a new name for the composite sample and then press ENT. You can now set the length of the samples you wish to merge.

If you have taken two samples, it could be that they are not the same length. For example, you may have a 2 second string sample and 3.5 second brass sample. If you were to merge these together, you would find that the string sample stops sounding before the brass sample. To overcome this, set the brass sample so that it is the same length as the string sample (i.e. set the number of sample points on the brass sample to be the same as the string sample) in the field marked 'last' next to the brass sample. The two samples will now be the same length. You can now press F8 (MIX) and the two samples will be merged together. To hear the result, press F1 (SLCT), select the new sample in the usual way and play it from the connected MIDI keyboard or controller or by pressing the ENT/PLAY button. If you find that the two samples are not ideally balanced, go back to the JOIN page (press F4 twice as above) and move the cursor to the relevant sample field - i.e adjacent to A or B and under where the display says 'scale'. You can now turn up or down (in dB's) the appropriate sample to balance the two together. This may be a bit of a trial and error routine but, as you have taken a copy of the two samples (i.e. in the new composite sample) you can't do any harm and you can repeat this procedure as often as you like, overwriting the unsatisfactory merged sample with each attempt. *(Note: if either of these samples was taken at full level, turning it up will result in distortion. Your only option, therefore, is to turn the other sample down. If both samples are already high in level, you will have to turn both down but one less than the other.)*

If you have not set the two samples to be the same length, you can change this afterwards by going to the TRIM page and editing the end point so that they both finish at the same time. Pressing CUT here will, of course, free up more memory.

You will note that the loop characteristics of both samples will be lost when you merge them so you will have to loop the new composite sample.

It is possible, of course, to repeat this procedure using the new composite sample as the basis of another. For example, you might like to layer the string and brass composite on top of a choir sample and this is done exactly as above except that the composite is used as one of the samples with the choir used as the other. This could be repeated again using, say, a tympani strike. The fact that the tympani only lasts a short time into this new composite is not important as you could loop the new sample at a point after the tympani has faded. As you can imagine, the possibilities are enormous.

NOTE: The resampling function cannot re-tune two differently tuned samples. For example, say you have a string sample taken at C3 and a brass sample taken at G3, even if you set the brass samples original pitch to C3 in ED2, the two samples will still be a fifth apart when merged. There is a way around this, however.

Using this as our example, go to ED2 and select RATE to resample one of the samples. Let us resample the brass sample. Select the brass sample in this page in the usual way and give it a new name (see the section on resampling for more details on this). Now move the cursor to the field marked NEW SAMPLE RATE and change the bandwidth until the field marked TUNE OFFSET reads -05.00 (in this example, assuming the original bandwidth is 44100 Hz, the new bandwidth should be 33050) and press GO (F8). Now go back to ED2 and set the new resampled brass samples original pitch to C3. Now go to ED1 and select JOIN and use the new resampled brass sample together with the original string sample to create your new merged sample name. Press MIX (F8) to create the merged sample and hear the results - the brass and the string sample will now be in unison.

This method can also be used to create detune effects by setting resampling rates that are not exact intervals (in this example, setting it so that the brass sample's tuning offset was -04.75 would have detuned the brass sample against the strings sample) and you could resample a sample in this way and then merge the resampled sample with the original to create lusher textures.

CHANGING SAMPLE LEVEL

This new sample merging facility also has another use and that is to change the volume of a single sample. Sometimes it is necessary to turn a samples level up because it was not sampled at a high enough level. Not only does this mean that certain samples don't balance up against other louder samples but, if a sample has not been taken at as a high a level as possible, your signal to noise ratio is effectively lower as you invariably have to crank up the gains on your mixer.

To adjust the level of a sample, select the sample you wish to affect in the field marked 'A :'. Now move the cursor to the field marked 'J :' and give it a new name name in the usual way by pressing NAME. Now set the amount you wish to change the level by in the SCALE filed and simply press F6 (A>J). The new sample named in J will be a copy of the original at a different level.

Note 1: If a sample was originally taken at too high a level and is distorted, turning the level of that sample will not reduce the distortion, it will simply make it quieter.

Note 2: If all this copying sounds tedious it is there for your own protection so that you won't lose any valuable sample data. There's nothing worse than taking a beautiful sample and then corrupting it somehow with no ability to recall the original. Copying may take a bit longer but it is worth it in the end. If this copying was not used, the only other option would be to keep having to save each sample to disk before editing it but this would take longer still.

LOOP TUNING

Sometimes, when you have set up a short, single cycle loop on a sample, you can find that whilst it is a perfect loop with no glitching and with a good sustain tone, it can be out of tune fractionally with the attack portion of the sound because the loop length is a multiple of an overtone in the sample that is not quite in tune with the rest of the sound. The new software offers an easy solution to this problem.

In ED2, there is a new facility called LOOP TUNE. This only affects loops that are held (i.e. loops with a value greater than 9998mS) and can be used to totally eliminate the effects described above. To use it, find a sound that displays this characteristic - a good solid attack that dies away to a very short, single cycle loop that is either sharp or flat. Now press F5 and you will see the new field on this page. Move the cursor to that field and move the data entry wheel until the loop is perfectly in tune with the attack. Although the tuning works in real-time (i.e. it changes as you change the data even though the note is held) you may find that you need to keep playing the note repeatedly to remind yourself of the attack portions tuning and to have some sort of tuning reference. Another way, of course, is to use a tuner. Tune the start of the sample before the loop until its perfectly in tune and then tune the loop against the tuner. In the absence of a tuner, a simple tone from a synth can be used as a reference.

Note: This facility can be used on very long loops as well and is not restricted to single cycle loops. If you find that you have a sample whose long loop is out of tune, experiment with this facility to rectify the problem.

That concludes our examination of the new sample editing functions of the Version 2.00 software. All other functions are unchanged so please refer to the original S1000 owners manual for more information on those.

NEW FUNCTIONS IN EDIT PROGRAM

VELOCITY SAMPLE START POINT - SMP3

There now exists a facility within Edit Program that allows you to use note-on velocity to dictate where a sample will start playing from. For example, you may have a 'cello sample with a heavy bowing in the attack. This might sound too obtrusive when played on every note but by using velocity sample start, you can set the program up so that you only hear the heavy bowing action with high note-on velocity. Used in conjunction with the velocity control of attack time in both ENV1 and ENV2, you can create extremely expressive instrumental samples.

This new facility exists in a new page called SMP3. Go into Edit Program and press F2 (KGRP). Now press SMP2 (F8). You will be presented with the usual page but you will also note that SMP2 has now changed to SMP3. By pressing F8 again, you will be given the new page that allows you to set this function up.

You can see that this facility is available on all 4 velocity zones independently giving you even greater control over your samples. Let's keep it simple, though, and use only zone 1. Assign your 'cello sample (or any sample that has a predominant attack) in SMP1 in the usual way and then get to SMP3. By setting a positive value (i.e. +9999), a heavy key velocity will start the sample right from the beginning and soft velocity will start the sample quite some way into it and missing the heavy attack. Setting a negative going value (i.e. -9999) will have the opposite effect. This latter option is not the usual but it can produce some interesting playing techniques. The scale, 0-9999 is simply an arbitrary numeric scale that allows you to govern the sensitivity of the velocity information received here. At low numeric levels (i.e. +0050) the sample will start a bit later on with low velocity input but not much. When set to higher numeric levels (i.e. +9999) the sample will start quite some way in. The scale does not relate to sample points directly.

Note 1: It is suggested that when you use this facility (especially with a positive going velocity start time in SMP3 such as +9999) that you set the attack time of ENV1 (and ENV2 if you are using this on the filter) to around 50 and set 'vel. > ATT' to around -20. This will set a velocity control of the envelopes attack time that is sympathetic to the velocity start time. Experiment!!

PROGRAM MASTER TUNE

Until now, the only way to tune a program 'globally' so that it was in tune with all the other programs was to individually tune the keygroups (although this is made easier by switching keygroup edit selection to 'ALL'). Now, in the renamed PTCH page (formerly PMOD) there is a new function called 'Tuning'. To tune a program, simply move the cursor to this field and adjust the tuning accordingly. You can adjust tuning in semitones or percentages of semitones for fine tuning.

INDIVIDUAL KEYGROUP VELOCITY SENSITIVITY

The new Version 2.00 software allows you to affect individual keygroups velocity sensitivity (before, this was a 'global' parameter that affected the whole program and all its keygroups). This can be especially useful with drums where you might want a fairly insensitive bass drum but a very sensitive snare drum for rolls and fills, etc..

The new function is available in the FILT page so, to access it, press KGRP (if you haven't already) and press FILT (F4). You will see the new field at the bottom left of the display. This allows you to increase or decrease the selected keygroups velocity sensitivity relative to the global velocity sensitivity set in the 'vel>loud' field of the OUT page. In other words, if the global velocity value is set to the default 20, if you set a value in your bass drum keygroup to -20, it will have no velocity sensitivity at all but if you also set your snare drum to +20 it will have far more dynamic range (in fact, it would be the equivalent of setting a global velocity value of 40 - i.e. (global) 20 + (individual) 20 = 40). Each and every keygroup can be set to be more or less velocity sensitive as you wish.

This facility can also be useful if you wish to increase the velocity sensitivity of the whole program but can't be bothered to go to the OUT page. Simply select ALL and set this field accordingly so that all keygroups' velocity sensitivity is adjusted.

KEY FOLLOW

Also new to the filter page is the ability to invert the keyboard tracking of the filters cutoff frequency and the new range is now -24 to +24. Although this is not something you will use every single day, it can be used very effectively, especially for special effects.

NEW LFO OPTIONS

One of the S950's most interesting program facilities is its ability to set different LFO parameters for each individual keygroup. This is not possible on the S1000 and cannot be changed. To overcome this, however, there are new options in the PTCH page of EDIT PROGRAM which allow you to control LFO speed, depth and delay using keyboard position. The range for each parameter is -50 to +50. It's primary use is to govern vibrato speed according to pitch. It is common, especially on strings, for vibrato to be slower on lower pitches and the vibrato for a 'cello is naturally slower than that on a violin. By setting, for example, a parameter value of, say, +25 on LFO speed for a strings program, you can more effectively emulate this. Similarly, you may like to set a parameter value of +15 for vibrato depth so that the high violins' vibrato will be slightly more intense than the basses and 'cellos. Likewise, setting a parameter value of +10 for LFO delay will mean that the higher pitched strings' vibrato will be introduced that much quicker than the lower pitched instruments - again, this is natural to 'the real thing' and will help make your string and other such sounds more realistic. When you play chords, the effect of these LFO's all running at slightly different rates and depths will make the sound much more vibrant and ensemble-like, especially with LFO DESYNC set to ON and so this technique can be equally effective on choir and vocal sounds, pad sounds, brass and so forth.

SETTING KEYSPAN USING A MIDI KEYBOARD

One of the most tedious aspects of making up your own programs on the old software was that keyspan for individual keygroups had to be set using the data wheel. This could be particularly tiresome in a complicated program that used a lot of keygroups. Now you can set the keygroups span using a MIDI keyboard (or any other MIDI controller).

To do this, assuming you have set up a good number of keygroups using the copy function in KGRP, go to the SPAN page of EDIT PROGRAM and place the cursor on the first LO field of KG1. Play a note on your keyboard - you will see that the display changes to show the note you played and the cursor automatically moved to the HI field. Now play the note you wish to have as the highest note for KG1 and you will see that this note is entered and the cursor has moved to the LO field of KG2. You can now repeat this procedure for all other keygroups. When you have inputted the highest note of the final keygroup, the cursor will automatically move back up to KG1.

NOTE: Be careful not to have the EDIT:ONE/ALL field set to ALL (unless you specifically want this) because as soon as you play a note to set a keygroup span, ALL the keygroups will have the same note assignment. This can be particularly infuriating if you have previously set up an elaborate program and you go to tweak one keygroup.

One tip, incidentally, when you are in the SPAN page that you may find useful. By placing the cursor on any KG, you can move through the keygroups simply by using the DATA wheel. This is easier and faster than moving though all the LO and HI fields and the tuning fields using the CURSOR wheel.

NEW DISPLAY IN SMP1

In software versions prior to Version 2.00, because the synth waveforms were assigned to the four velocity zones in SMP1, the display looked a bit confusing. Now, because the synth waveforms are not used, they do not appear in SMP1 resulting in a much clearer screen display making it much easier to assign samples and set up velocity switching and crossfades.

NOTE: Should you wish to clear unused samples from unused velocity zones in existing programs, you may like to try this:

1. *Go into EDIT PROGRAM, press KGRP and select SMP1.*
2. *Move the cursor to the ONE/ALL field and select ALL.*
3. *Now move the cursor to the first of the unused velocity zones (i.e. zone 2 or 3 - whatever) and press NAME. Now fill that name with blank spaces by pressing the -/> button until a totally blank space is created. Press ENTER.*
4. *Move the cursor to the next unused velocity zone and repeat the procedure given in 3.*
5. *Repeat procedure 3 for the next unused velocity zone if there is one.*
6. *Repeat this procedure for other programs.*
7. *Resave these programs to disk.*

What you have done here is fool the S1000. You have typed in a name (even though it comprises only blank spaces, the S1000 thinks it's a sample name) and this has the effect of creating a clearer screen. Because you selected ALL, this has affected all the keygroups in the program. This has no effect on the program other than creating a clearer display to work with should you wish to edit the program at some point.

CHANGING THE LEVELS OF INDIVIDUAL KEYGROUPS

We have received many comments about the fact that it is difficult to change the levels of individual keygroups so as to be able to balance them against each other, especially in drum programs. The main complaint was about increasing level rather than turning keygroups down.

The reason that it was not always possible to turn keygroups up in level a great deal was due to the default value selected in the OUT page for overall level. In order to optimise signal to noise ratios, a fairly high value was used (90). Whilst giving a 'hotter' overall output level for the S1000, it left little room to turn keygroups up in level. Now the default for this parameter has been changed to 80 giving you better control over level for individual keygroups. The only down side is that the S1000's overall output level is marginally quieter and so you may have to increase it's output level control or tweak the gains or channel fadeers of your mixer up fractionally. We feel this is preferable so that you can balance keygroups up more efficiently.

If you wish to increase the overall output level of the S1000, simply go to the OUT page and change the LOUDNESS field to 90 but remember that you will restrict your ability to balance keygroups against each other.

That concludes our examination of new routines in EDIT PROGRAM. We hope you find them useful and beneficial to your work.

ASSIGNING AUTOMATIC HARD DISK LOADS USING MIDI PROGRAM CHANGE

In a live situation, it is often desirable, even necessary, to be able to load up sounds and programs into your S1000 remotely. This can now be done if you possess a hard disk drive (either the S1000HD or an external hard disk drive using a suitable interface). This 'auto-load' function can also be useful when you are sequencing as it allows you to call up sounds in advance as the sequencer is running simply by inserting a suitable MIDI program change command in the sequence at an appropriate moment (remembering, of course, that you will need at least 12 seconds for a full 2 meg memory load, longer if memory expansion boards are fitted!!).

To assign a MIDI program number to a hard disk volume, go to the disk page by pressing the main DISK button and select the volume you wish to number. Now press REN (F3). This function is used to name a volume but now it also has a new field - 'vol load number:' and this is where you assign a MIDI program number which will activate a full disk load. Simply assign the program number of your choice in the field and then press F7 (VOL). Now, whenever you send out that MIDI program number to the S1000, that volume will automatically load. Repeat this procedure for any other volumes you need to load in this way. Remember, of course, that volumes you do not need to load in this way need not be given a number. With careful 'housekeeping' you can organise your live set so that sounds can be quickly and easily loaded from your master keyboard or sequencer automatically without any disruption of the gig.

Note 1: This new feature is NOT a 'load while play' function. It will activate the disk drive in the usual way, preventing access to any sounds during the load procedure. Load while play is still being investigated by our designers but there are certain problems with it that need to be overcome before it is released.

Note 2: Be careful that the program numbers you assign to a hard disk volume don't coincide with a program number you may wish to use WITHIN a volume. For example, if you assign MIDI program number 3 to a volume and that volume brings up program number 1 when it has loaded into the S1000, if you then try to call up program 3 within that volume using MIDI program change, you will activate the automatic hard disk load function again!!

Note 3: If you do not use a hard disk of any description, this function is inoperative. You cannot use it to activate a floppy disk load.

FORMATTING HARD DISK DRIVES LARGER THAN 60 MEG

It is now possible to use hard disk drives whose storage capacity is greater than 60 Megabytes. This is done using a system of partitions. If, for example, you have a hard disk of 600 Meg, you would format the disk to be in partitions of 60 Meg - you would end up with 10 partitions of 60 Meg and each partition is readily accessible.

To do this, simply select the format option in DISK and the screen will display the new field HARD PARTITIONS. It is here you select the size of your partitions which can be 30, 40, 50 or 60 Meg. You cannot mix and match different partitions to be different sizes and the size selected here will affect all partitions.

To select a partition, you will see that the main disk display now shows:

HARD:A

A partition is automatically assigned an alphabetical character and you can move the cursor to the character field to select the appropriate partition. From there you can select volumes in the usual way in any of the partitions. This new software retains total compatibility between hard disks formatted on older software versions.

This new software allows you to use such peripheral equipment as the new Sony Magneto Optical Disk, CD ROM and so forth. Please check with your local Akai dealer for information on compatible peripheral equipment.

LOADING S900 DISKS

Version 2 allows you to load samples AND programs from S900 or S950 disks. Simply insert the disk, press GO or CLR and GO and the S1000 will load the sounds. It will still go through the usual "Unscrambling S900 data" procedure so it can take a little extra time but you can resave the data as S1000 data so that this will not occur next time you load that disk but remember that if you overwrite the original S900 disk, you will not be able to use it with an S900.

*NOTE 1: You will notice that S900 programs have a * symbol after them. This is normal. They will be loaded sequentially and program numbers will be automatically assigned within the S1000.*

NOTE 2: Most S900 information is retained and recognised by the S1000, especially the all important stuff like span, envelopes, filter settings, etc., but individual output assignments are not because of substantial hard and software differences between the two instruments. S900 programs that have individual keygroups set to different MIDI channels for multi-timbral sequencing will not retain this when loaded into the S1000 as this facility cannot exist on the S1000. As a result, you will have to redefine such disks within the S1000 so that each sound is in a different program on its own MIDI channel. Similarly, some confusion exists within the S1000 regarding the ONE SHOT function of the S900 as it cannot differentiate between the two. It looks at the one-shot selection as PLAY TO SAMPLE END even if this is selected in EDIT PROGRAM on the original S900 disks. As a result, samples placed in programs that use the ONE SHOT mode will play to sample end. This can be easily changed in the S1000 either by editing the individual samples or changing the PLAYBACK field in SMP2 of EDIT PROG.

NEW DELETE OPTIONS

In keeping with the new delete functions in Version 2.00, if you wish to delete anything from disk, hard or soft, the S1000 checks with you to make sure you want to proceed.

SAVING DEFUALT PARAMETERS AND SCSI ID NUMBERS

If you have your new software on disk, it is possible to save your sample editing working defaults within the new software. For example, if you have the IB104 Digital Interface fitted and you only sample through that and not through the front panel, you could set REC1 to default to this. Similarly, you may set your own default record times, bandwidth, sampling start modes, stereo or mono, timestretch parameters, resampling rate, etc., and then save those to your Systems floppy so whenever you boot up your Version 2.00 software, your chosen working defaults are recalled.

Similarly, it is possible to save your external hard disks SCSI ID number as part of the software.

To do this, turn the S1000 on with the Version 2.00 software in the disk drive. The S1000 will load the new software automatically. Now set your sampling defaults and SCSI ID's accordingly. Now go to the DISK mode and press SAVE and select OPERATING SYSTEM. By resaving the Version 2.00 (either onto the existing disk or onto a new disk for safety's sake), whenever you boot up the Version 2.00 it will load up with your sampling defaults and SCSI ID number without having to set this up every time.

NOTE: This feature only works with disk based software. It cannot be done if your software is installed in ROM. You will note also that if you have Version 2.00 on ROM but try to boot up from power-on with a floppy, the internal ROM softwrae takes priority. You can, however, load the operating software in the DISK mode after you have turned the S1000 on with ROM Version 2.00 and this will override the ROMs.

NEW MIDI RECEIVE MODE

Not much has changed in this mode except that now it is possible to set the MIDI channel for the MIDI receive monitor. This can be especially useful if you are sequencing multi-timbrally and need to see what is going on. To use this feature press MIDI and then press RCVE. The display now has the option for you to select the channel you wish to look at.

At the top right of the display is the new field marked CHAN: and this can be set to OMNI or on individual channels 1-16. You may now selectively look at individual MIDI channels being received by the S1000. This setting also affects the PPM display given when F3 is selected.

One more new field is now on F8 - KBD. This is only for use with the new S1000KB and relates to keyboard options on that instrument. You will receive the display:

no internal keyboard

if you press this button.

NEW OPTIONS IN MASTER TUNE

There are now a few extra new facilities in MASTER TUNE that relate to the use of a soft pedal (i.e. MIDI controller #67). To access these new functions press MASTER TUNE and then pressF3 - SOFT.

You will see that you may now use this MIDI controller to control LOUDNESS REDUCTION, ATTACK STRETCH and FILTER CUTOFF.

Basically, whenever the soft pedal is used, the following will happen:

LOUDNESS REDUCTION - a value set here will allow you to reduce the volume whenever the soft pedal is used.

ATTACK STRETCH - a value set here will allow you to lengthen the attack time of the envelope generators, thereby creating a softer sound when you press the soft pedal.

FILTER CLOSE - pressing the soft pedal will close the cutoff frequency of the filter by an amount set here. This too will create a softer sound.

These new facilities will help you create more expression with your S1000 especially when used in conjunction with other MIDI controllers.

FURTHER INFORMATION REGARDING THE S1000

USING THE INDIVIDUAL OUTPUTS AND THE STEREO OUTPUTS

We get many 'phone calls and letters concerning certain functions within the S1000 that seem to be causing problems for some reason. The one area which seem to cause the biggest problem is assigning things to the individual outputs.

You can assign two things to any individual output - whole programs and/or individual keygroups. Assigning a whole program to an individual output can be done in two ways. Firstly, you can do it in SELECT PROGRAM by pressing F3 (RESP) and using that page to assign any given program to any given output or the same thing can be done in the OUT page of EDIT PROGRAM. This output assignment flexibility is most useful if you are sequencing multi-timbrally.

Let us say that you have 8 programs which you want to sequence multi-timbrally. Firstly, give ALL the programs the same program number by pressing F2 (RNUM) and assigning a the same program number to all the programs. Now press F8 (GO) and this will renumber the programs. All the programs are now layered on top of each other. You can now press F3 (RESP) and assign each program its own MIDI channel (i.e. 1-8 in this example although they could be anything you choose relevant to your particular circumstances). Now, by moving the cursor to the field marked 'Mono out' each program can be assigned to its own output for individual processing on a mixer. *(Note: At this point, all the programs should also be appearing out of the stereo outputs as well but we will look at this later.)*

All should be well - each program should be coming out of its own specified individual output. But you can run into problems here. You see, it is possible to assign individual keygroups to individual outputs as well. This is done in EDIT PROGRAM in SMP2 of KGRP mode. There is a group of fields (one for each velocity zone for each keygroup) that allows you to assign a keygroup zone to an individual output using a system of offsets. For example, if your drum program is globally assigned to output 1, you can have the bass drum coming out of output 1 by setting no offset. By setting the snare drum keygroup to an offset of +1, this will come out of individual output 2 (i.e. global output 1 + 1 (offset) = output 2). Similarly, the four toms could be offset by +2, +3, +4 and +5 to appear at outputs 3, 4, 5 and 6 respectively. Cymbals and other drum voices can be assigned the same way. It appears that this is where some confusion lies. It is possible, for example, to load up a drums program with keygroups assigned to individual outputs as described above. Then you may load in a piano program and assign that program to output 2. Then you load up a bass program and assign that to output 3. What happens is that the piano is sharing the same output as the snare and the bass is sharing the same output as one of the toms. The only way around this is to re-program the set-up using a mixture of EDIT PROGRAM (to access the individual keygroups' assignments) and the RESP page of SELECT PROGRAM or the OUT page of EDIT PROGRAM.

Another source of confusion arises when someone tries to assign, say, a strings program to output 4 but finds it comes out of output 6. This is because that program for some reason has an offset of +2 and so any global output assignment is offset by two outputs.

Should you run into these problems, you may find all this of some help in sorting out any confusion that may arise. The key area to investigate if you experience problems like this is the 'out' field in SMP2 of EDIT **PROGRAM as it's likely that this is where things are going awry.**

Whilst this may be a source of confusion (and even experienced S1000 programmers have been known to scratch their heads over this) it does allow for a great deal of flexibility. Further flexibility is afforded on the S1000 as far as individual output assignment is concerned because it is possible to not only assign programs or keygroups to individual outputs but to also use the stereo outs at the same time.

You could for example, have all your programs mixed out of the stereo outputs in a complimentary mix and also have the various programs assigned to individual outs. The stereo mix would be fine for working at home where you maybe don't have a large mixer. Then, when your song is all worked out, you can take the S1000 to the studio, connect the individual outs and mix the instruments properly with the right EQ, effects and so forth.

In another example, you may have 8 programs assigned to individual outs. By mixing these completely out of the stereo outputs (using the 'stereo level' field in the RESP page of SELECT PROG) you may now include another program which is not assigned to any of the individual outputs but is mixed into the stereo outs. Furthermore, you could include other programs which are not assigned to an individual output but only appear at the stereo outs. You can also use the stereo outs as two extra individual outputs. You could have 8 programs assigned to the 8 outputs. Another program could be mixed only into the stereo output (but with no individual output assignment) and panned hard left. Another program, again with no individual output assignment, could be mixed into the stereo and panned hard right - this would have the effect of these latter programs effectively having their own outputs, one coming solely out of the left, the other only out of the right.

Want to add yet another individual output to make a total of eleven individual outs??

Assign the first 8 programs to outputs 1-8. Assign the other two to the left and right outputs using the pan controls as described above. Now add another program - don't assign it to any individual output and mix it right out of the stereo outputs. Now switch the field marked 'effect output' in the RESP page to 'ON'. Now, you have eight programs coming out of individual outputs, two more programs coming out of the left and right outputs respectively and one more program coming out of the effect send output - 11 programs out of 11 individual outputs.

Of course, because all of these outputs are totally polyphonic, other programs can be added that share outputs so even more flexibility is there.

It is hoped that this will give you some assistance should you run into any confusion with individual outputs. It is also hoped that you have gained some ideas for output assignment beyond that which is given in the manual.

Let us now move on to sampling drums.

By far the easiest way to sample drums is to sample everything on C3. You can now create a drum program in EDIT PROGRAM with as many keygroups as necessary assigned to the required notes. Now go to SMP1 and assign the drums accordingly. At first, you will hear the drums being played at the wrong pitches so, whilst still in SMP1, switch the keygroup edit selection from ONE to ALL and move the cursor to the field marked 'pitch'. It should read TRACK for each of the 4 zones. Change that using the data knob so that it reads CONST. By having selected ALL, all keygroups will be changed simultaneously.

What you have done is to turn the keyboard off so it doesn't matter where you play on the keyboard, the notes will always sound at C3 which, of course, is the note you sampled the drums on. Using this method, you can insert any drum from any disk or other program and place it in any other program on any note and it will always be at the correct pitch. Of course, you can tune individual drums up and down using SMP2. In this way, you need only ever create one general drum and/or percussion program and use that as a template for all other drum programs you wish to create.

EDITING SAMPLE START TIMES AND DEALING WITH DELAYS

The S1000 automatically finds an optimum start time when you create a new sample. Basically, it searches for the most significant level at the start of the attack. Sometimes, especially with drums, that start time may not be as precise as it should be and flamming can occur when drums are played simultaneously. To rectify this, we suggest that you edit the start point manually using the visual editing display in ED1 (TRIM) of EDIT SAMPLE. This way, you can edit the start times so that drums start precisely on the attack, eliminating any flamming effects. To help you ensure that your percussive sounds start dead on, we suggest that you maybe play the sound several octaves down - this will accentuate any delays during note-on and the actual start of the sample. At the same time, it will also show up any distortion that may exist from taking a sample at too high a level.

Another source of timing discrepancies arises when you change the a value or change screens whilst the S1000 is being played. This is because the software is running 'screen refresher' routines - as soon as the display is changed in any way, the processor has to use some if its power to deal with these screen changes and so timing can be disrupted. This is normal and your S1000 does not have a fault.

Another source of delays can arise when you try to layer several stereo samples on top of each other. For example, when trying to create that 'killer' snare, you may try to layer two or three different stereo snares. Unfortunately, owing to the speed restrictions of MIDI and the processing time required to ensure total stereo phase coherence, there can sometimes be minute delays introduced. Sadly, there is not much that can be done about this and we suggest you experiment to discover alternatives. With the new software, of course, your killer snare could be created by merging several drum samples together using SAMPLE MERGE in EDIT SAMPLE so that the various component sounds become one big stereo sample. This will render no reduction in polyphony as well as ensuring accurate timing and phase coherence.

We hope you find this information useful. Of course, as always, if you have any problems, please contact your local Akai dealer or distributor. Similarly, if you have any ideas for future updates, please let us know and we will see if they are able to be included in future software revisions.